IP Phone User Manual

V.06

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1 Introduction

This user's manual is for Altesys IP Phone, which will explain the keypad instruction, web configuration, and command line configuration. Before using the IP Phone, some setup processes are required to make the IP Phone work properly. Please refer to the Setup Menu for further information.

1.1 Hardware Overview

The IP Phone has the following interfaces for Networking, telephone interface, LED indication, and power connector.
Two RJ-45 Networking interface, these two interfaces support 10/100Mps Fast Ethernet. you can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.

1.2 Software Overview

Network Protocol	**Tone**
• SIP v1 (RFC2543), v2(RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/ PPPoE Client • Telnet/HTTP Server • DNS Client	• Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • User Programming Tone
	Phone Function
Codec	• Volume Adjustment • Speed dial, Phone book • Flash • Speaker Phone
• G.711: 64k bit/s (PCM) • G.723.1: 6.3k / 5.3k bit/s • G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: adds VAD & CNG to G.729	**IP Assignment**
	• Static IP • DHCP • PPPoE
Voice Quality	**Security**
• VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	• HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
Call Function	• ToS field
• Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference	**NAT Traversal**
	• STUN
DTMF Function	**Configuration**
• In-Band DTMF • Out-of Band DTMF • SIP Info	• Web Browser • Console/Telnet • Keypad
SIP Server	**Firmware Upgrade**
• Registrar Server (three SIP account) • Outbound Proxy	• TFTP • Console • HTTP • FTP

2 Keypad interface for IP Phone demo system

2.1 Keypad description

Key Name	Description	Note
1	“1”, “-“, “,”, “!”, “?”	
2	“2”, “a”, “b”, “c”, “A”, “B”, “C”	
3	“3”, “d”, “e”, ”f”, “D”, “E”, “F”	
4	“4”, “g”, “h”, “I”, “G”, “H”, “I”	
5	“5”, “j”, “k”, “l”, “J”, “K”, “L”	
6	“6”, “m”, “n”, “o”, “M”, “N”, “O”	
7	“7”, “p”, “q”, “r”, “s”, “P”, “Q”, “R”, ‘S”	
8	“8”, “t”, “u”, “v”, “T”, “U”, “V”	
9	“9”, “w”, “x”, “y”, “z”, “W”, “X”, “Y”, “Z”	
0	“0”, “space”	
*	“*”, “•”, “:”, “@”	
# / Send	Start dialing process	
Transfer	This is “Transfer” to the other phone number	
REDIAL	This is “REDIAL” the same number again	
HOLD	This is “HOLD” function	
Del/Mute	This is “Cancel” (cancel settings) and “Mute” function	
DND	This is “Reject” function	
ENTER	This is “OK”, accept setting	
Phone Book	This is the “Phone book” menu	
UP/DOWN	This is Up↑ and Down↓ key	
FWD	This is “Forward” function	
MENU	This is the “Menu” key to set the IP Phone	
H-F	This the Speaker Phone	
Pick Up	This is the “Pick Up” function	
Volume -/+	This is volume setting	
Conf	This is three way conference function	
Call Log	This is “Call Log” list menu	

2.2 Keypad Function and setting List

Phone Book

2.2.1.1 Search:Search Phone Book.
2.2.1.2 Add entry:Add new phone number to phone book.
2.2.1.3 Speed dial:Add speed dial phone number to speed dial list.
2.2.1.4 Erase all:Erase all phone number from Phone Book.

Call Log

2.2.1.5 Incoming calls: Show all incoming call.
2.2.1.6 Dialed numbers: Show all dialed call.
2.2.1.7 Erase record: Delete call history.

1. All: Delete all call history.
2. Incoming: Delete all incoming call.
3. Dialed: Delete all dialed out call.

Phone setting

2.2.1.8 Call forward

2.2.1.8.1 All Forward.
1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.2.1.8.2 Busy Forward.
1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.2.1.8.3 No Answer Forward.
1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.2.1.8.4 Ring Timeout: Set the Ring times to start the no answer forward function, ex: 2 means after 2 rings then forward to the dedicated number.

2.2.1.9 Do not disturb

1. All:
2. By Time:
3. Duration: Set the start time and end time to Block Setting.

2.2.1.10 Date/Time setting: Date and Time Setting.

2.2.1.10.1 Date & Time: Set the IP Phone Date and Time.
2.2.1.10.2 SNTP setting
2.2.1.10.2.1 SNTP : Enabled / Disable SNTP.
2.2.1.10.2.2 Primary SNTP: Set Primary SNTP server IP address.
2.2.1.10.2.3 Secondary SNTP: Set Secondary SNTP server IP address.
2.2.1.10.2.4 Time zone: Set Time zone.
2.2.1.10.2.5 Adjustment Time: Set adjustment time period.

2.2.1.11 Volume and Gain

2.2.1.11.1 Handset volume: Set Handset volume from 0~15 (max.) for you to hear.
2.2.1.11.2 Speaker volume: Set Spearer phone volume from 0~15 (max.) for you to hear.
2.2.1.11.3 Handset Gain: Set Handset Gain from 0~15 (max.) for the other site to haer.
2.2.1.11.4 Speaker Gain: Set Spearer phone Gain from 0~15 (max.) for the other site to haer.

2.2.1.12 Ringer

2.2.1.12.1 Ringer volume: Ringer volume setting from 0~15 (max.).
2.2.1.12.2 Ringer type: Ringer tone selection from 1~4.

2.2.1.13 Auto Dial: Set Auto Dial time from 3~9 seconds.

2.2.1.14 Auto Answer: Set Auto Answr for user can re-dial a call from IP call to PSTN call or from PSTN call to IP call. This function will active on IP Phone with FXO interface.

2.2.1.15 Answer Counter: Set Auto Answer will active after the numbers of ring. This function will active on IP Phone with FXO interface.

Network

2.2.1.16 General

2.2.1.16.1 IP Type

1. Fixed IP client
2. DHCP client
3. PPPoE client

2.2.1.16.2 Fixed IP setting

2.2.1.16.2.1 Host IP
2.2.1.16.2.2 Network mask
2.2.1.16.2.3 Gateway IP
2.2.1.16.2.4 MAC address

2.2.1.16.3 PPPoE setting

2.2.1.16.3.1 User name
2.2.1.16.3.2 Password

2.2.1.16.4 DNS Server

2.2.1.16.4.1 Primary DNS
2.2.1.16.4.2 Secondary DNS

2.2.1.17 Status: Show IP address and MAC address.

SIP Settings

If you want to use Keypad to set the SIP setting, you have to go to item 7 (Administrator) System Authent to input the password, or you can not change the SIP setting.

2.2.1.18 Service domain

2.2.1.18.1 First realm

2.2.1.18.1.1 Activation
2.2.1.18.1.2 User name
2.2.1.18.1.3 Display name
2.2.1.18.1.4 Register name
2.2.1.18.1.5 egister password
2.2.1.18.1.6 Proxy server
2.2.1.18.1.7 Domain server
2.2.1.18.1.8 Outbound proxy

2.2.1.18.2 Second realm

2.2.1.18.2.1 Activation
2.2.1.18.2.2 User name
2.2.1.18.2.3 Display name
2.2.1.18.2.4 Register name
2.2.1.18.2.5 Register password
2.2.1.18.2.6 Proxy server Proxy
2.2.1.18.2.7 Domain server
2.2.1.18.2.8 Outbound proxy

2.2.1.18.3 Third realm
2.2.1.18.3.1 Activation
2.2.1.18.3.2 User name
2.2.1.18.3.3 Display name SIP
2.2.1.18.3.4 Register name
2.2.1.18.3.5 Register password
2.2.1.18.3.6 Proxy server Proxy
2.2.1.18.3.7 Domain server
2.2.1.18.3.8 Outbound proxy

2.2.1.19 Codec
2.2.1.19.1 Codec type
1. G.711 uLaw
2. G.711 aLaw
3. G.723
4. G.729
5. G.726-16
6. G.726-24
7. G.726-32
8. G.726-40

2.2.1.19.2 VAD

2.2.1.20 RTP setting
2.2.1.20.1 Outband DTMF
2.2.1.20.2 Duplicate RTP
1. No duplicate
2. One duplicate
3. Two duplicate

2.2.1.21 RPort Setting: RPort Enabled/Disabled

2.2.1.22 Hold by RFC

2.2.1.23 Status: Show the SIP Proxy register status. You can use UP/Down key to check each Realm's status.
1. First Realm
2. Second Realm
3. Third Realm

NAT Transversal
2.2.1.24 STUN setting
2.2.1.24.1 STUN
2.2.1.24.2 STUN server

Administrator

2.2.1.25 Auto Config

2.2.1.25.1 Config Mode: You can select Disable/TFTP/FTP to do the auto config function. This function must work with the Auto Config Server.

2.2.1.25.2 TFTP server: Setting the TFTP server IP address.

2.2.1.25.3 FTP server: Setting the FTP server IP address.

2.2.1.25.4 FTP Login Name: Setting the login name to the FTP server.

2.2.1.25.5 FTP Password: Setting the Password to the FTP server.

2.2.1.26 Default setting: You can restore to the default setting..

2.2.1.27 System Authentication: To do the SIP setting from Keypad, need to input the password first. Default is “test”.

2.2.1.28 Version: This will show the system’s firmware version.

2.2.1.29 Watch Dog: You can use this to enable Watch Dog function to do the debugging.

2.2.1.30 Restart: You can use this function to restart your IP Phone.

3 Setup the IP Phone by Web Browser

The IP Phone provides a built-in web server. You can use Web browser to configure the IP Phone. First please input the IP address in the Web page. In the end of IP address, please add the port number ":9999". Ex:http://192.168.123.1:9999

3.1 Login.

Please input the username and password into the blank field. The default setting is:

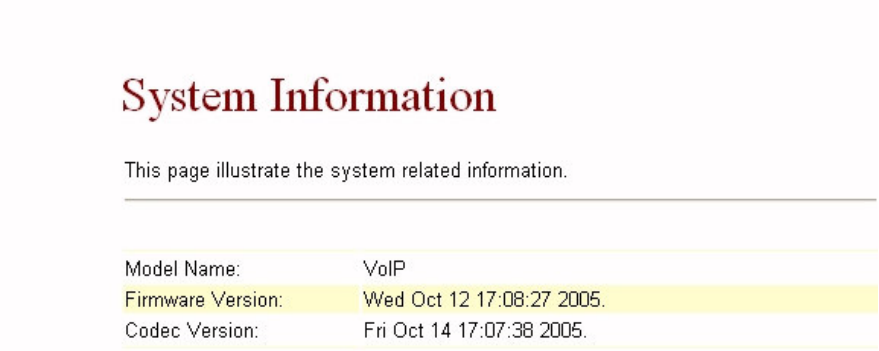
1. For Administrator, the username is: root and password is: test. If you use the account login, you can configure all the setting.
2. For normal user, the username is: system or user; and the password is: test. If you use the account login, but you can not configure the SIP setting.

Click the "Login" button will move into the IP Phone web based management information page.

If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.

3.2 System Information for the IP Phone.

When you login the web page, you can see the IP Phone current system information like firmware version, company… etc in this page.
Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.


System Information
This page illustrate the system related information.
Model Name: VoIP
Firmware Version: Wed Oct 12 17:08:27 2005.
Codec Version: Fri Oct 14 17:07:38 2005.

3.3 Phone Book

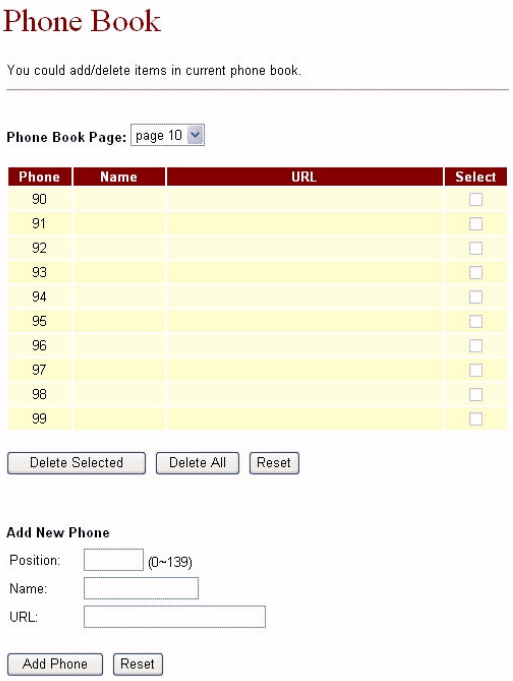
In Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial number. The Phone Book can store 140 phone numbers and the Speed Dial can store 10 phone numbers. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press “#”.

In the Phone Book function you can add/delete the phone number in the phone book list. You can input maximum 100 entries phone book list.

3.3.1.1 If you need to add a phone number into the phone book, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

3.3.1.2 If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

3.3.1.3 If you want to delete all phone numbers, you can click “Delete All” button.

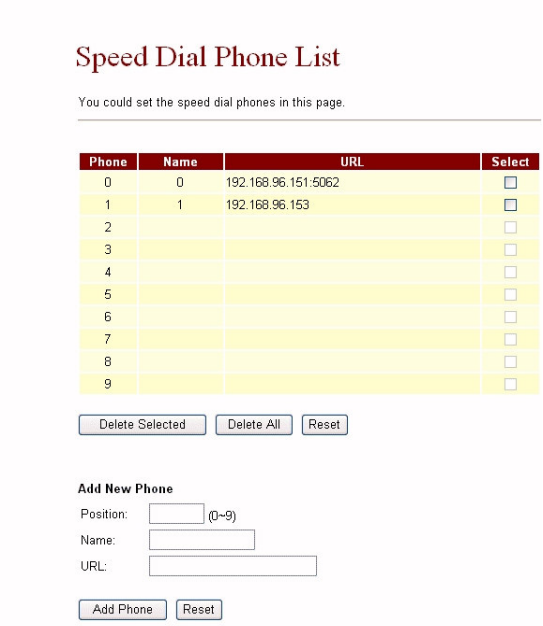


In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

3.3.1.4 If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

3.3.1.5 If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

3.3.1.6 If you want to delete all phone numbers, you can click “Delete All” button.



3.4 Phone Setting

In Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Melody Settings, Block Setting, and Auto Dial Setting functions.

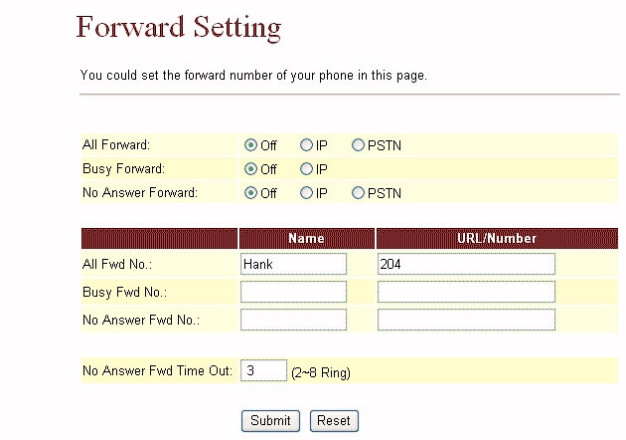
Call Forward function: you can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

3.4.1.1 All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

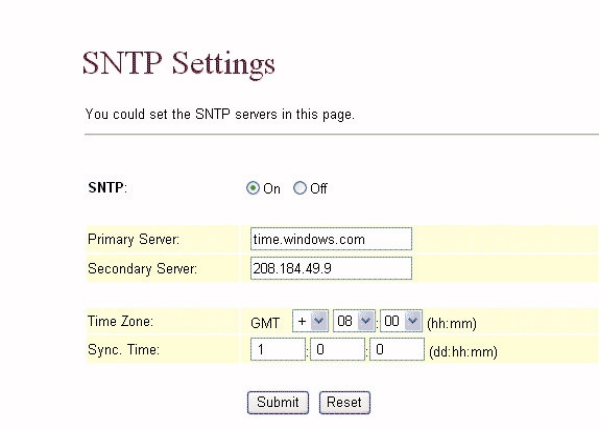
3.4.1.2 Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

3.4.1.3 No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

3.4.1.4 When you finished the setting, please click the Submit button.

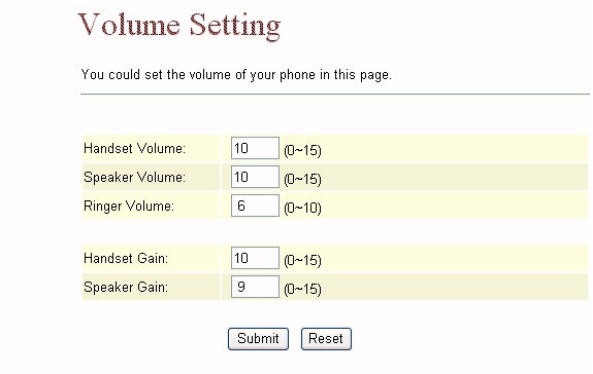


SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

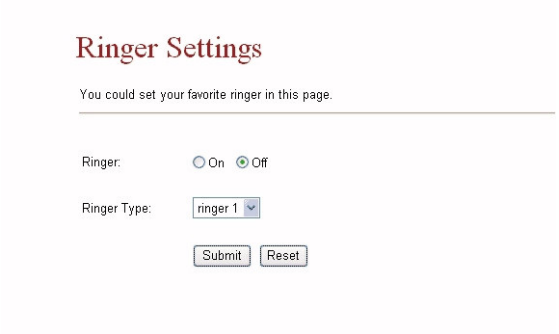


Volume Setting function: you can setup the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

3.4.1.5 Handset Volume is to set the volume you hear from the handset.
3.4.1.6 Speaker Volume is to set the volume you hear from the speaker phone.
3.4.1.7 Ringer Volume is to set the ringer volume.
3.4.1.8 Handset Gain is to set the volume send out from from the handset.
3.4.1.9 Speaker Gain is to set the volume send out from from the micro phone.



Melody Setting function: you can select the melody for the imcoming call. When you finished the setting, please click the Submit button.

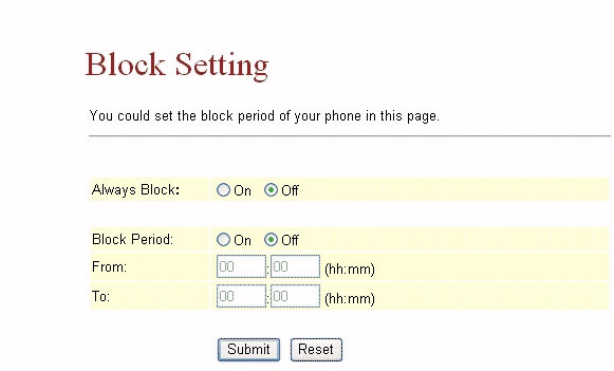


Block Setting function: you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

3.4.1.10 Always Block: All incoming call will be blocked until disable this feature.

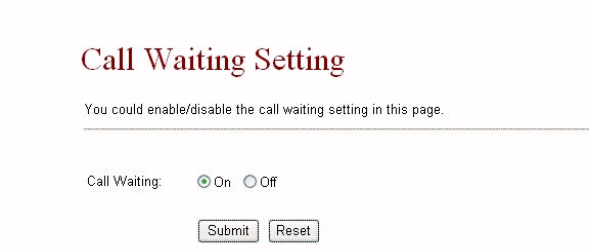
3.4.1.11 Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

3.4.1.12 When you finished the setting, please click the Submit button.


Block Setting
You could set the block period of your phone in this page.
Always Block: On Off
Block Period: On Off
From: 00 : 00 (hh:mm)
To: 00 : 00 (hh:mm)
Submit Reset

Auto Dial Setting function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

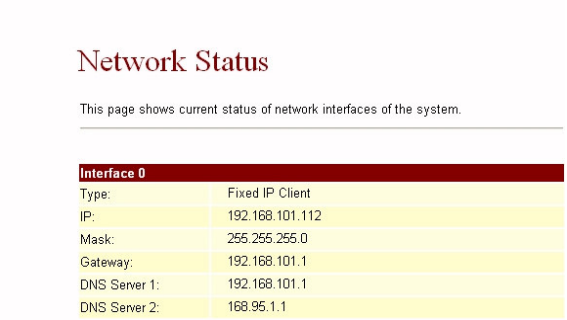
Call Waiting Setting function: If user doesn't want to be inform there is a new incoming call, user can set the function off.



3.5 Network

In Network you can check the Network status, configure the Network Settings and DDNS settings.

Network Status: You can check the current Network setting in this page.



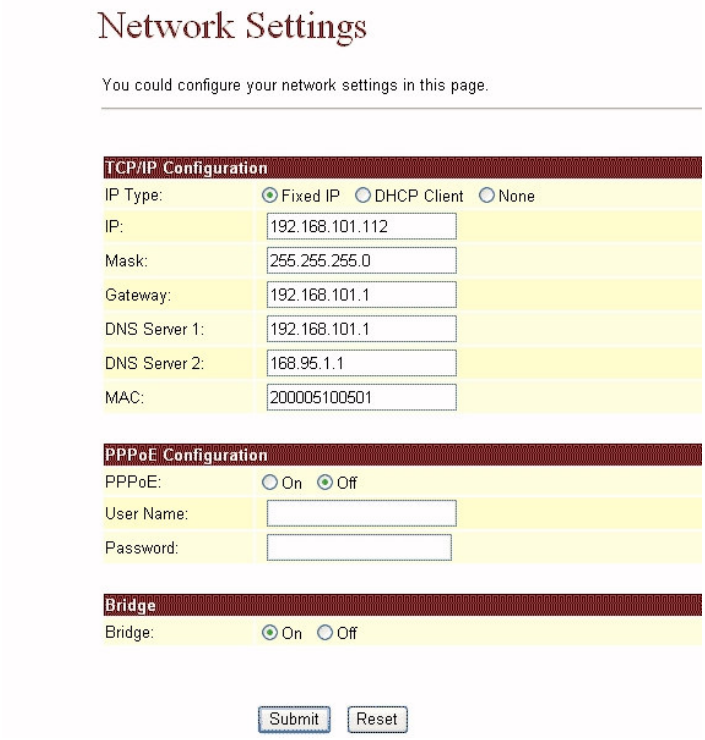
Network Settings: You can configure the IP Phone Network setting in this page.

3.5.1.1 The TCP/IP Configuration item is to setup the LAN port's network environment. You may refer to your current network environment to configure the IP Phone properly.

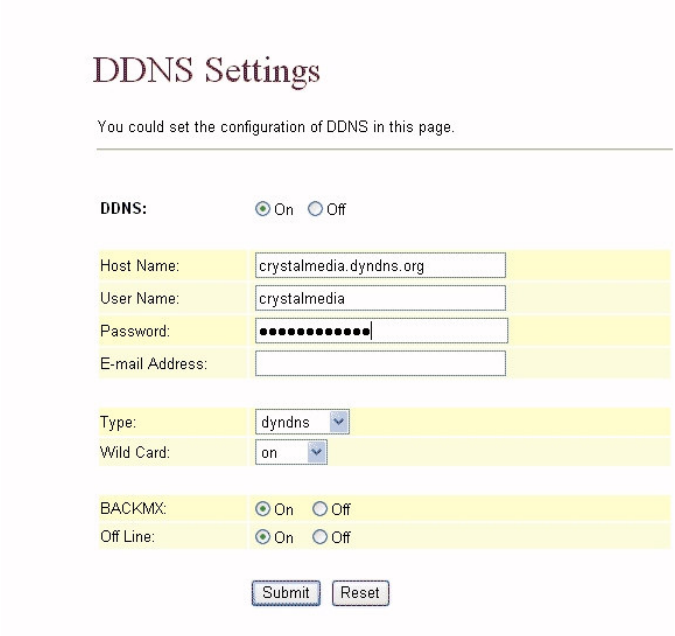
3.5.1.2 The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

3.5.1.3 The Bridge Item is to setup the IP Phone Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.

3.5.1.4 When you finished the setting, please click the Submit button.



DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.



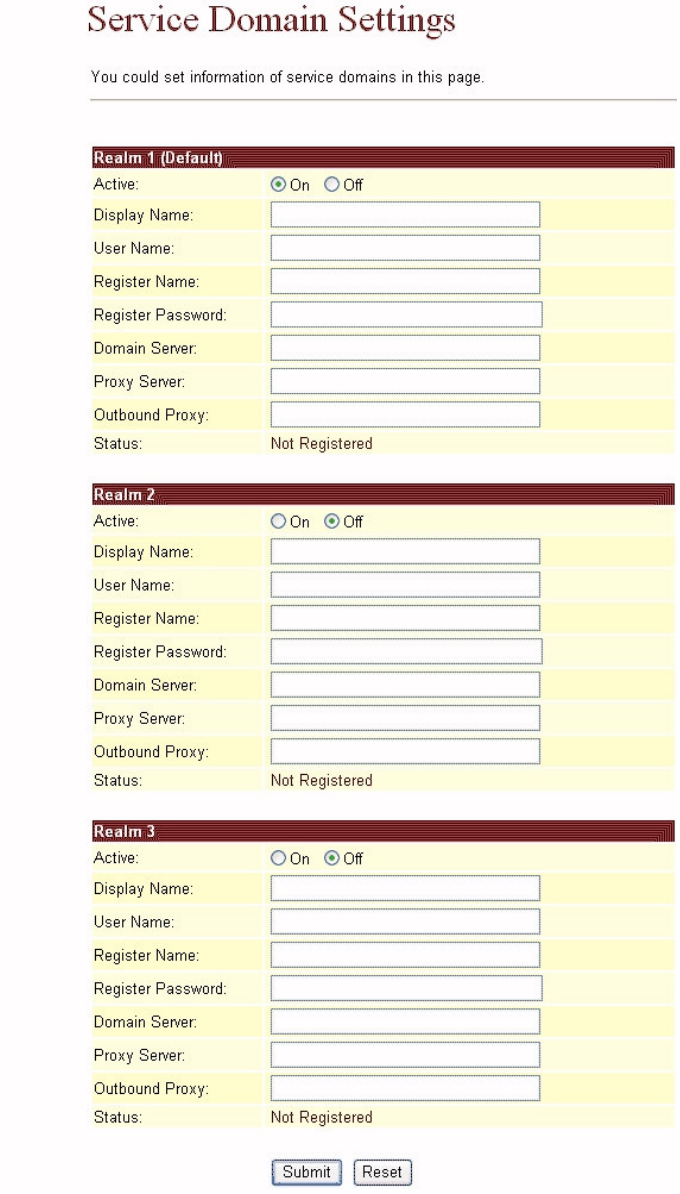
3.6 SIP Settings

In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related informations correctly then you can register to the SIP Proxy Server correctly.

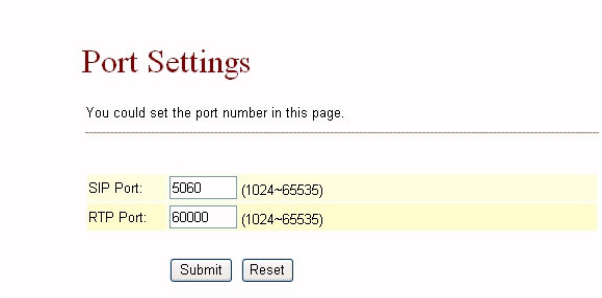
In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the IP Phone. You can dial the IP Phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

3.6.1.1 First you need click Active to enable the Service Domain, then you can input the following items:

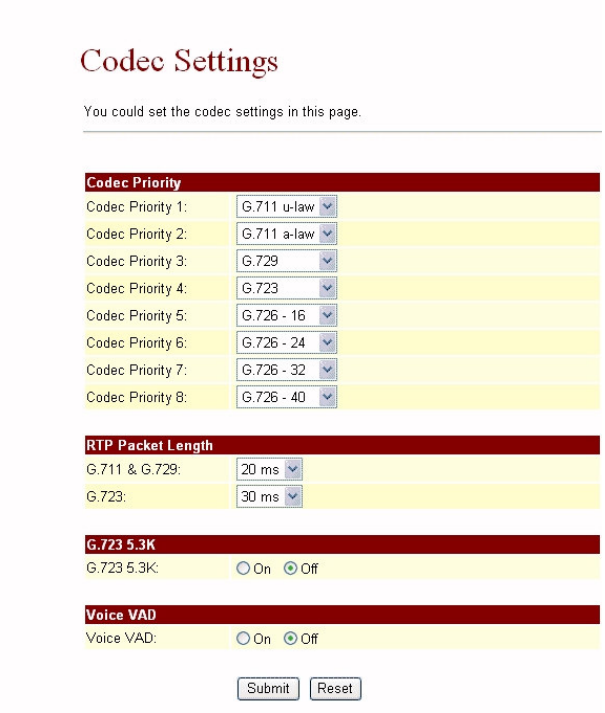
- 3.6.1.1.1 Display Name: you can input the name you want to display.
- 3.6.1.1.2 User Name: you need to input the User Name get from your ISP.
- 3.6.1.1.3 Register Name: you need to input the Register Name get from your ISP.
- 3.6.1.1.4 Register Password: you need to input the Register Password get from your ISP.
- 3.6.1.1.5 Domain Server: you need to input the Domain Server get from your ISP.
- 3.6.1.1.6 Proxy Server: you need to input the Proxy Server get from your ISP.
- 3.6.1.1.7 Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.
- 3.6.1.1.8 You can see the Register Status in the Status item. If the item shows “Registered”, then your IP Phone is registered to the ISP, you can make a phone call direcly.
- 3.6.1.1.9 If you have more than one SIP account, you can following the steps to register to the other ISP.
- 3.6.1.1.10 When you finished the setting, please click the Submit button.



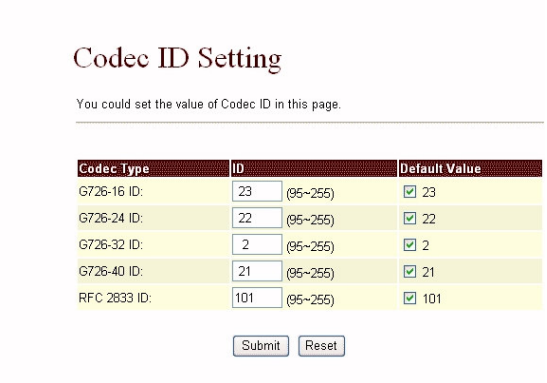
Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.



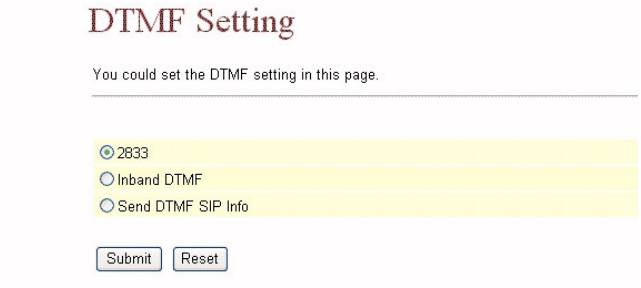
Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.


Codec Settings
You could set the codec settings in this page.
Codec Priority
Codec Priority 1: G.711 u-law
Codec Priority 2: G.711 a-law
Codec Priority 3: G.729
Codec Priority 4: G.723
Codec Priority 5: G.726 - 16
Codec Priority 6: G.726 - 24
Codec Priority 7: G.726 - 32
Codec Priority 8: G.726 - 40
RTP Packet Length
G.711 & G.729: 20 ms
G.723: 30 ms
G.723 5.3K
G.723 5.3K: On Off
Voice VAD
Voice VAD: On Off
Submit
Reset

Codec ID Settings: you can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.

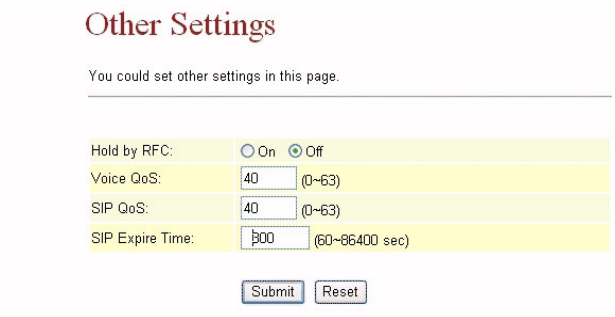


DTMF Setting: you can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



RPort Function: you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

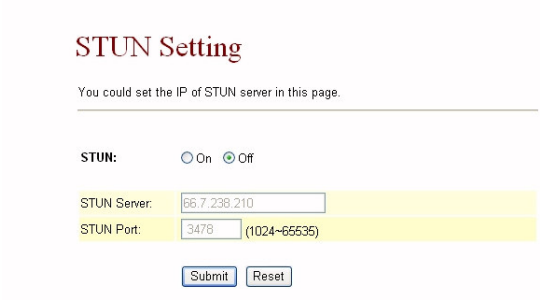
Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.



3.7 NAT Trans.

In NAT Trans. you can setup STUN function. These functions can help your IP Phone working properly behind NAT.

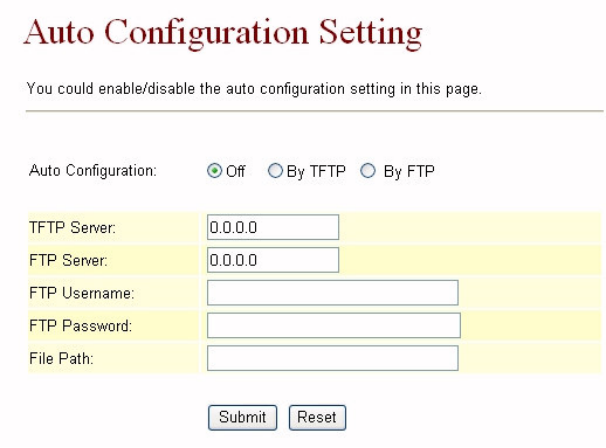
STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your IP Phone working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.



3.8 Others.

In Others you can setup Auto Config and ICMP Setting function. The function can configure your IP Phone automatically.

Auto Config: you can setup the Auto Configuration Enable/Disable and auto configuration by FTP or TFTP. You need to select the way to do the Auto Configurationand set the Server IP address in this page. This function can automatically download the configure file to setup your IP Phone. When you finished the setting, please click the Submit button.



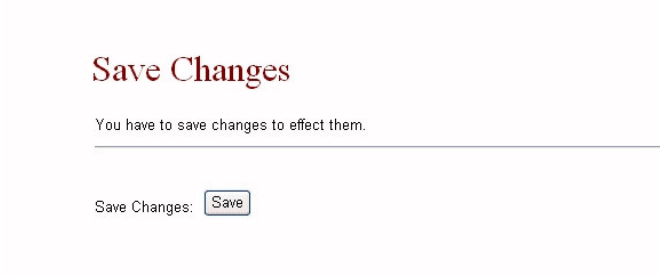
ICMP Setting: you can setup the ICMP echo Enable/Disable in this page. This function can disable echo when someone ping this device, it can avoid haker try to attack the device. When you finished the setting, please click the Submit button.

3.9 System Auth.

In System Authority you can change your login name and password.

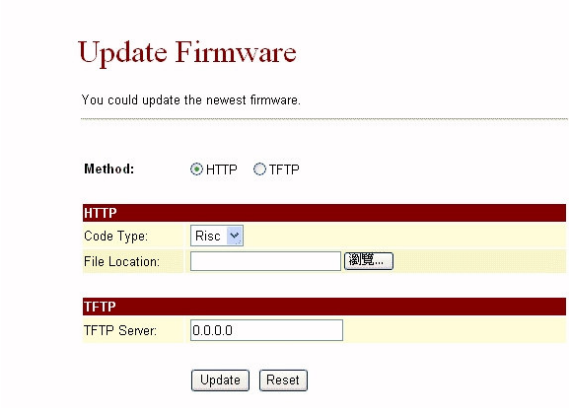
3.10 Save Change

In Save Change you can save the changes you have done. If you want to use new setting in the IP Phone, You have to click the Save button. After you click the Save button, the IP Phone will automatically restart and the new setting will effect.



3.11 Update

In Update you can update the IP Phone's firmware to the new one or do the factory reset to let the IP Phone back to default setting.
In New Firmware function you can update new firmware via HTTP in this page. You can ugrade the firmware by the following steps:
3.11.1.1 Select the firmware code type, Risc or DSP code.
3.11.1.2 Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
3.11.1.3 Select the correct file you want to download to the IP Phone then click the Update button.



In Default Setting you can restore the IP Phone to factory default in this page. You can just click the Restore button, then the IP Phone will restore to default and automatically restart again.

Restore Default Settings
You could click the restore button to restore the factory settings.
Restore default settings:
Restore

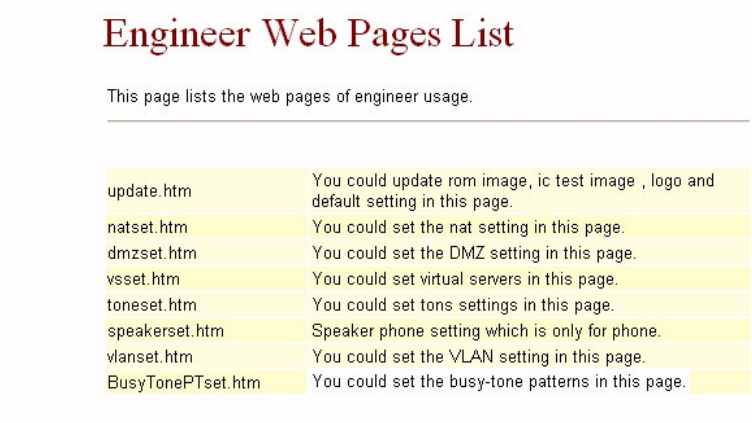
3.12 Reboot

Reboot function you can restart the IP Phone. If you want to restart the IP Phone, you can just click the Reboot button, then the IP Phone will automatically.

4 Engineering webpage

4.1 Engineer usage webpage list

You have to login the system first then change the webpage manually. In this webpage you will see the list about engineer webpage. You can change the webpage to what you want.

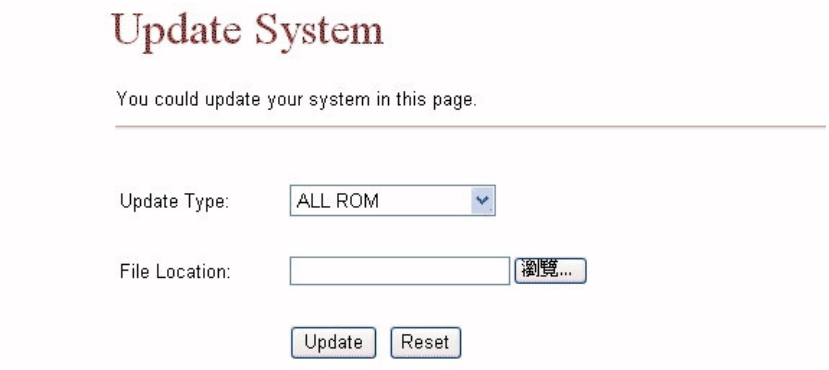


Engineer Web Pages List

This page lists the web pages of engineer usage.

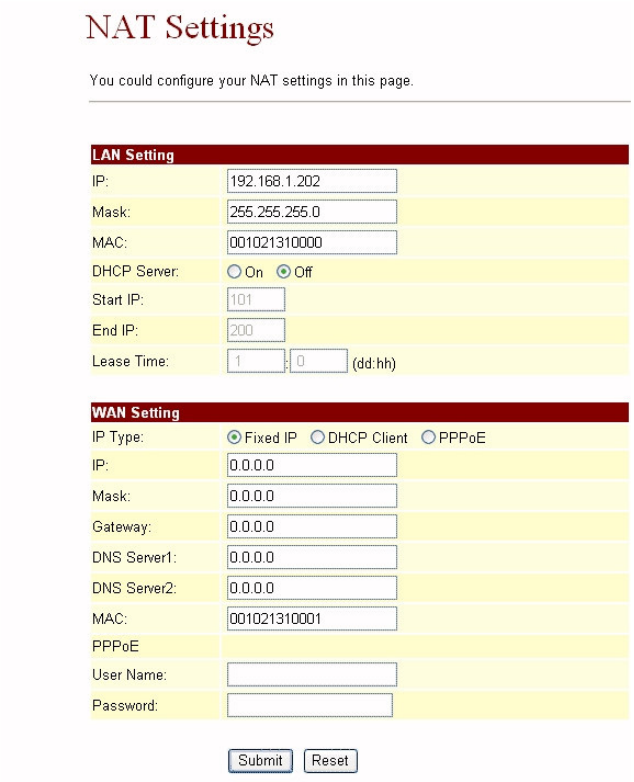
update.htm	You could update rom image, ic test image , logo and default setting in this page.
natset.htm	You could set the nat setting in this page.
dmzset.htm	You could set the DMZ setting in this page.
vsset.htm	You could set virtual servers in this page.
toneset.htm	You could set tons settings in this page.
speakerset.htm	Speaker phone setting which is only for phone.
vlanset.htm	You could set the VLAN setting in this page.
BusyTonePTset.htm	You could set the busy-tone patterns in this page.

Update.htm
You have to login the system first then change the webpage to update.htm manually (http://ip address:9999/update.htm).
In this page you can update the system's ROM code, IC Test , Default setting and Logo.
Update ROM code. You can update the ROM code from this function. Please be noted that if you update the wrong file or during the update process the power is off, the system will be crashed.
Update Logo. The Logo specification is 220x170 jepg file.



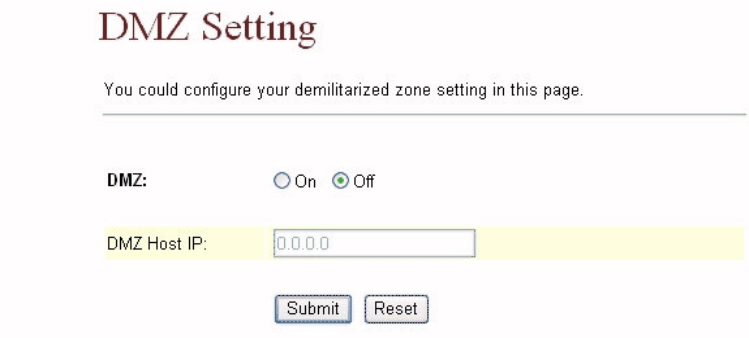
4.2 Natset.htm

You have to login the system first then change the webpage to natset.htm manually (http://ip address:9999/natset.htm).
In this page you can setup the nat function. The WAN setting is for you to set how the get the IP address for the device. The LAN setting is for the other devices to get the IP address from the device. You can choose to use DHCP server or not.



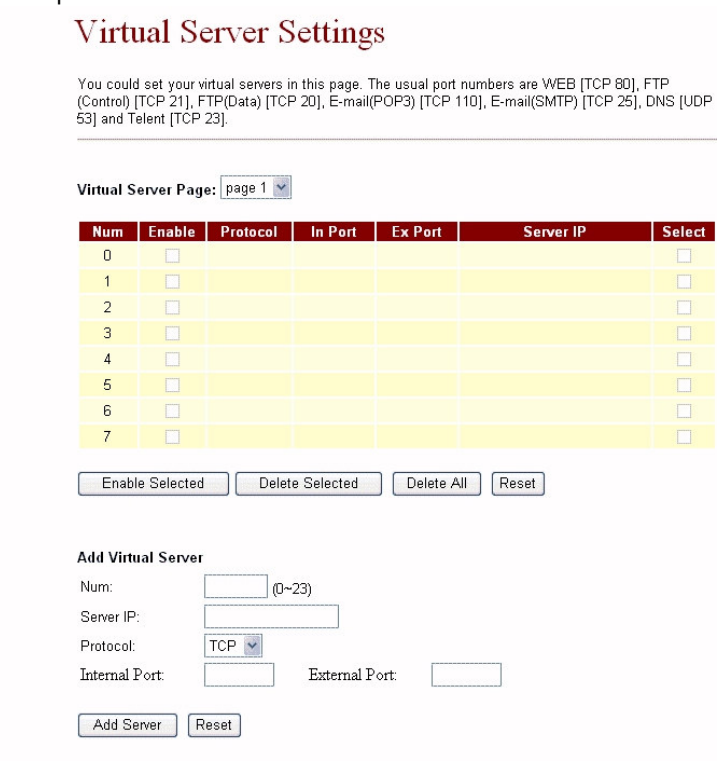
4.3 Dmzset.htm

You have to login the system first then change the webpage to dmzset.htm manually (http://ip address:9999/dmzset.htm).
In this page you can setup the DMZ function. You need to enable/disable this function and set the IP address for DMZ.



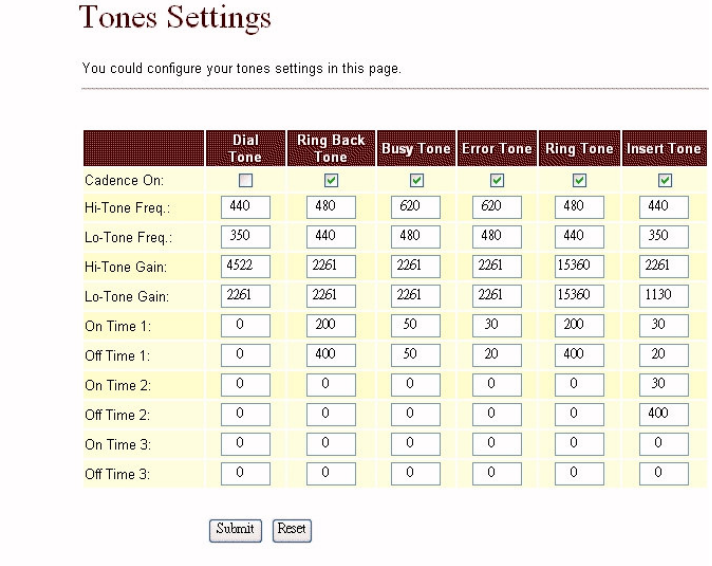
4.4 Vsset.htm

You have to login the system first then change the webpage to natset.htm manually (http://ip address:9999/vsset.htm).
In this page you can setup the Virture Server function.


Virtual Server Settings
You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].
Virtual Server Page: page 1
Num Enable Protocol In Port Ex Port Server IP Select
0 1 2 3 4 5 6 7
Enable Selected
Delete Selected
Delete All
Reset
Add Virtual Server
Num: (0~23)
Server IP:
Protocol: TCP
Internal Port:
External Port:
Add Server
Reset

4.5 Toneset.htm

You have to login the system first then change the webpage to natset.htm manually (http://ip address:9999/toneset.htm).
In this page you can setup the Tone frequency and cadence to meet the requirement.

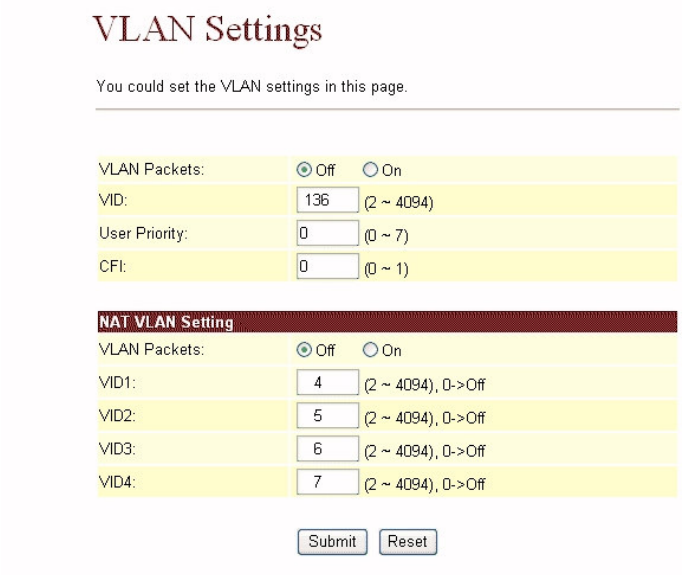

Tones Settings
You could configure your tones settings in this page.
Dial Tone
Ring Back Tone
Busy Tone
Error Tone
Ring Tone
Insert Tone
Cadence On:
Hi-Tone Freq.: 440 480 620 620 480 440
Lo-Tone Freq.: 350 440 480 480 440 350
Hi-Tone Gain: 4522 2261 2261 2261 15360 2261
Lo-Tone Gain: 2261 2261 2261 2261 15360 1130
On Time 1: 0 200 50 30 200 30
Off Time 1: 0 400 50 20 400 20
On Time 2: 0 0 0 0 0 30
Off Time 2: 0 0 0 0 0 400
On Time 3: 0 0 0 0 0 0
Off Time 3: 0 0 0 0 0 0
Submit
Reset

4.6 Speakerset.htm

You have to login the system first then change the webpage to natset.htm manually (http://ip address:9999/toneset.htm).
In this page you can setup the Speaker function. Default we set the speaker is in half-duplex mode. If you want to set to full-duplex mode, it need to check if your housing is suitable for this function. Or you have to set it as half-duplex mode.

4.7 Vlanset.htm

You have to login the system first then change the webpage to natset.htm manually (http://ip address:9999/vlanset.htm).
In this page you can setup the VLAN function.
VLAN Packets: if enable then the device will only accept packet with VLAN Tag and the Tag must be the same as VID you set.
VID: You can follow your service provider to set your VID.
User Priority: Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.
CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.
NAT Mode: When you set your device in NAT mode, you can separate the other device connectd behind your device into 4 group. You can set different VID for these 4 group.


VLAN Settings
You could set the VLAN settings in this page.
VLAN Packets: Off On
VID: 136 (2 ~ 4094)
User Priority: 0 (0 ~ 7)
CFI: 0 (0 ~ 1)
NAT VLAN Setting
VLAN Packets: Off On
VID1: 4 (2 ~ 4094), 0->Off
VID2: 5 (2 ~ 4094), 0->Off
VID3: 6 (2 ~ 4094), 0->Off
VID4: 7 (2 ~ 4094), 0->Off
Submit Reset

5 Phone function list

When your IP Phone is configured properly, you can make a phone call to other IP Phone located in the same service provider. If you want to make a phone call, you can dial the phone number then press "#" button to start to dial the phone number or wait for a while then system will dial the number automatically.

The IP Phone provides some functions that list as below:

1. Call Hold: You can push the Hold key to hold the current call for a while, then push Hold key again to keep talking.

2. Call Waiting: When a new call is coming while you are talking, you can push the Hold button to switch to the new call. You can push the Flash button to switch between the two calls.

3. 3-Way Conference: If you want to make a 3-way conference call, you can make a phone call to the first phone number. After the call is established, push the Hold button then you can hear the Dial tone, then make a phone call to the second phone number. When the second call is established, press CONF button.

4. Call Transfer: Current we can support 3 kind of transfer application. Below is the operation method.

   A. Normal Transfer: A call B then transfer to C
   Step 1: A call B
   Step 2: B press "Transfer" then A will be held
   Step 3: Make a new call to C
   Step 4: After C answer the call then Hang Up to complete transfer

   B. 2 way Transfer: A call B then transfer to C but C is busy or C reject transfer then B want the call back to A
   Step 1: A call B
   Step 2: B press "Transfer" then A will be held
   Step 3: Make a new call to C
   Step 4: If C is busy or reject then B hang up the phone
   Step 5: Press "Hold" to restore call with A

   C. Blind Transfer: A call B then transfer to C and do not care about C's situation
   Step 1: A call B
   Step 2: B press "Transfer" then A will be held
   Step 3: Make a new call to C
   Step 4: B hang up the phone

5. Redial: User can push the Redial button to dial the last dialed number.

6. Flash: User can push the Flash button to make the IP Phone to dial mode.

7. Speaker Phone: You can use Speaker phone to make a phone call.

8. Pre Dial: User can dial the number first, after finished then raise the handset or push the speaker button; the IP Phone will start to dial.

6 Get a FWD account

1. The website is www.freeworlddialup.com; you can apply an account to use the VoIP communication. You can follow the instruction to input the information. After you finished, you will receive a mail sent by the FWD mail system, you will get the account information in the mail.

2. When you got the account, you can setup the related information into the IP Phone.
3. You can setup the related information into the IP Phone by web browser. You need to input the Proxy Name, Domain Name, Register Name, and password. The Display Name you can input what you want to let others see.
4. After you registered to the SIP Server, you can try to call your friends who also registered in the same SIP Server. You just need to dial your friend's user name (registered name) and press "#" then you can make a phone call to your friend.
5. If you want to make a phone call to the other in the internet, first you need to registered in a Proxy Server (with SIP Server IP, Domain IP, registered name, Password), make sure you already enable Stun function, then you can try